PATENT

Please delete the last three paragraphs on page 6, line 24 and substitute the following paragraph therefor:

In one embodiment, the present invention provides a method for manipulating a received sound signal to produce a sound signal. The received sound signal is received from a packet-switched network that looses some packets. In one step, a first received frame that is part of the received sound signal is received. A first signal frame corresponding to the first received frame is produced. The first signal frame is part of the sound signal. A second received frame is normally produced contiguously with the first received frame. During production of the first signal frame, it is determined that part of the second received frame is currently unavailable for production. An expanded portion is produced after determining that the part of the second received frame is currently unavailable. The first signal frame and the expanded portion are contiguous parts of the sound signal. The expanded portion corresponds to a different amount of the received sound signal than either the first or second received frames. Various aspects of the present invention are defined in the appended claims.

A2

IN THE CLAIMS:

Please cancel claims 1-12 without prejudice to or disclaimer of the subject matter contained therein.

B

A5

13. (Amended) The method of claim 26, wherein use is made of an oscillator model for extracting signal segments from the first signal frame, the oscillator model including a codebook in which vectors of samples forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

14. (Amended) The method of claim 13, wherein the second-listed producing step comprises a step of matching a true state of a trailing part of the first signal frame with said states in said codebook, and reading out a signal segment from said codebook that corresponds to the state having been matched with said true state.

PATENT

A3
B

15. (Amended) The method of claim 13, wherein said signal segments of said codebook have variable lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the expanded portion to a consecutive signal frame.

16. (As Filed) The method of claim 13, wherein time delays between said states in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

17. (As Filed) The method of claim 14, wherein the states and the corresponding segments of said codebook are scaled in order to improve the matching with said true state.

18. (As Filed) The method of claim 14, wherein merging of said true state is performed with the matching state of said codebook.

B

19. (Amended) The method of claim 14, wherein the second-listed producing step involves performing the corresponding operations with respect to a heading part of the second signal frame being consecutive to the expanded portion.

A4

20. (Amended) The method of claim 26, wherein said first signal frame is either a sound signal frame resulting from a complete decoding operation of the first received frame, or an intermediate time-domain signal frame resulting from a partial decoding operation of the first received frame.

21. (Amended) The method of claim 26, including the step of using an oscillator model, which oscillator model includes a codebook in which vectors of samples of a received digitized sound signal forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

Please delete claims 22-25 without disclaimer of or prejudice to the subject matter contained therein.

PATENT

26. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

B

receiving a first received frame that is part of the received sound signal; 3:45 - 4:41

producing a first signal frame corresponding to the first received frame, 5:45 - 6:56 wherein:

the first signal frame is part of the sound signal, and

a second received frame is normally produced contiguously with the first received frame;

determining after beginning the first-listed producing step that at least part of the second received frame is currently unavailable for production; and *3:45-47 Covell

producing an expanded portion after the determining step, wherein: 4:55 - 67

A5

the first signal frame and the expanded portion are contiguous parts of the sound signal, and 4:55 - 67 6:27 - 36 9:29 - 57

the expanded portion that corresponds to a different amount of the received sound signal than either the first or second received frames. 4:55 - 67

27. (New) The method of claim 26, wherein the expanded portion is selected from the first signal frame based, at least in part, upon measures of periodicity. Covell 9:29-57

28. (New) The method of claim 26, wherein the determining step comprises a step of determining near the end of production of the first signal frame if the part of the second received frame is currently unavailable for production. 3:45 - 4:41 5:45 - 6:56

29. (New) The method of claim 26, further comprising steps of:

determining after beginning the second-listed producing step that the second received frame is still unavailable for production; 3:45 - 4:41 5:45 - 6:56

producing a second expanded portion after the immediately-preceding determining step, wherein the expanded portion and the second expanded portion are contiguous parts of the sound signal. Covell

30. (New) The method of claim 26, wherein:

a playback time of the expanded portion is variable, and Covell

the playback time is selected based, at least in part, upon the sound signal. Covell

B

31. (New) The method of claim 26, wherein:

the first signal frame includes a plurality of sound samples, and

the expanded portion is determined with a time resolution finer than a sample rate of the plurality of sound samples.

32. (New) The method of claim 26, further comprising a step of producing a second expanded portion based, at least in part, on some of the second received frame, wherein the expanded portion and second expanded portion are contiguous parts of the sound signal.

A5

33. (New) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal using a periodicity measure, whereby any audible discontinuities between the expanded portion and second expanded portion are reduced.

34. (New) The method of claim 26, wherein the signal frame corresponds to a plurality of received frames.

35. (New) The method of claim 26, further comprising a step of merging the expanded portion and a contiguous, subsequent, portion of the sound signal based, at least in part, on overlap-add, wherein a time shift of the first signal frame and expanded portion is optimized based, at least in part, on correlation.

36. (New) The method of claim 26, further comprising steps of:

measuring overload of a jitter buffer;

discarding some of the second received frame based, at least in part, on the overload; and

merging a preceding and a subsequent portions of the sound signal after the discarding step.

B

37. (New) The method of claim 36, further comprising steps of:

determining if a signal fitting criteria between the preceding and subsequent portions is fulfilled; and

performing the discarding step only with the immediately-preceding determining step is fulfilled.

38. (New) The method of claim 36, wherein a length of the some of the second received frame is based, at least in part, on the sound signal.

39. (New) The method of claim 36, wherein the some of the second received frame comprises a plurality of sub-portions that are sequentially discarded.

~~A5~~

40. (New) The method of claim 36, wherein:

the merging step is based, at least in part, on overlap-add, and

any time-shift of the preceding and subsequent portions is optimized based, at least in part, on a measure of periodicity.

41. (New) A computer-readable medium having computer-executable instructions for performing the computer-implementable method of claim 26.

42. (New) A computer system adapted to perform the computer-implementable method of claim 26.

43. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein the first signal frame is part of the sound signal;

determining after beginning the first-listed producing step that part of the second received frame currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the expanded portion and a second signal frame are contiguous parts of the sound signal,

the first signal frame and the second signal frame would be contiguous parts of the sound signal in situations where the part of the second received frame is available for production, and

the expanded portion is a different size than either the first or second received frames.

B

44. (New) A method for manipulating a received sound signal to produce a sound signal, wherein the received sound signal is received from a packet-switched network that looses some packets, the method comprising steps of:

receiving a first received frame that is part of the received sound signal;

producing a first signal frame corresponding to the first received frame, wherein:

the first signal frame is part of the sound signal, and

a second received frame is produced contiguously with the first received frame when the second received frame is available;

determining after beginning the first-listed producing step that part of the second received frame is currently unavailable for production; and

producing an expanded portion after the determining step, wherein:

the first signal frame and the expanded portion are contiguous parts of the sound signal,

the expanded portion replaces at least some of the second received frame, and

the expanded portion is a different size than either the first or second received frames.

A5

REMARKS

Attached hereto is a marked-up version of the changes made to the specification and claims by the current amendment. The attached page is captioned **"Version with marking to show changes made."**

Amendments

Claims 1-12 and 22-25 have been canceled. New Claims 26-44 have been added. Therefore, claims 13-21 and 26-44 are present for examination. Portions of the summary are amended according to the claims in their present form. These amendments are fully supported by the specification and add no new matter.

35 U.S.C. §102 Rejection, Shlomot

The Office Action has rejected former claims 1-5, 20 and 22-25 under 35 U.S.C. §102(e) as being anticipated by U.S. Patent No. 6,377,931 issued to Shlomot ("Shlomot"). Applicants believe amended claims 13-21 and 26-44 are not anticipated by Shlomot as limitations in these claims are neither taught nor suggested. More specifically, Shlomot only deals with whole frames when compressing or expanding, whereas the claimed invention operates on frame portions. As can be appreciated by those skilled in the art, operating upon portions of frames allows for expansion and compression with higher quality of service.

Shlomot only teaches compression or expansion with whole segments or frames. See Shlomot col. 6, lines 53-61, and col. 7, lines 5-20. For example, three segments could be compressed into one segment, or two segments could be expanded into three segments. These received segments are fixed to a predetermined size. See Id. col. 4, lines 4-10. Each received segment corresponds one-to-one with the size of played segments. See Id. col. 4, lines 51-54, and col. 6, lines 3-9. Therefore, Shlomot cannot manipulate portions of segments or frames. For at least this reason, Applicants believe Shlomot cannot anticipate the amended claims.

35 U.S.C. §103 Rejections

The Office Action has variously rejected some dependent under 35 U.S.C. §103(a) as being unpatentable over Shlomot in view of some other references. Applicants note that Shlomot only contemplates variations in the timing of received packets and not the complete loss of packets. Accordingly, smoothing is of little concern in Shlomot. It is unclear why one of ordinary skill in the art would modify Shlomot to